In the Claims:

Please amend the claims as follows:

1. (currently amended) A method ~~of filtering a speech signal~~ comprising: providing a filter suited for reduction of distortion, including quantization noise, caused by speech coding of a speech signal; estimating background acoustic noise in said speech signal; adapting said filter in response to the estimated background acoustic noise to obtain an adapted filter; and applying said adapted filter to said speech signal so as to reduce background acoustic noise and to reduce distortion, including quantization noise, caused by speech coding in said speech signal.

2. (previously presented) The method as defined in claim 1, wherein said adapting said filter involves adjusting filter coefficients of said filter.

3. (previously presented) The method as defined in claim 2, wherein said estimating, adapting and applying are performed for portions of said speech signal which contain speech as well as for portions which do not contain speech.

4. (previously presented) The method as defined in claim 2, wherein said filter includes a short-term filter function designed for attenuation between spectrum formant peaks of said speech signal and wherein said filter coefficients include at least one coefficient that controls the frequency response of said short-term filter function.

5. (previously presented) The method as defined in claim 4, wherein said filter includes a spectrum tilt compensation function and wherein said filter coefficients include at least one coefficient that controls said spectrum tilt compensation function.

6. (previously presented) The method as defined in claim 1, wherein background acoustic noise in said speech signal is estimated as relative noise energy and noise spectrum tilt.

7. (previously presented) The method as defined in claim 2, wherein said adapting is performed by selecting values for said filter coefficients from a lookup table, which maps a plurality of values of estimated background acoustic noise to a plurality of filter coefficient values.

8. (previously presented) The method as defined in claim 1, wherein said estimating, adapting and applying are performed after decoding said speech signal.

9. (previously presented) The method as defined in claim 1, wherein estimating, adapting and applying are performed before encoding said speech signal.

10. (previously presented) The method as defined in claim 1, wherein said speech signal comprises speech frames and wherein said estimating, adapting and applying are performed on a frame-by-frame basis.

11. (previously presented) The method as defined in claim 7, further comprising initially generating said lookup table by: adding different artificial noise power spectra having given parameter (s) of background acoustic noise to different clean speech power spectra; optimizing a predetermined distortion measure by applying said filter to different combinations of clean speech power spectra and artificial noise power spectra; and for said different combinations, saving in said lookup table those filter coefficient values, for which said predetermined distortion measure is optimal, together with corresponding value (s) of said given parameter (s) of background acoustic noise.

12. (previously presented) The method as defined in claim 11, wherein said predetermined distortion measure includes spectral distortion.

13. (previously presented) The method as defined in claim 11, wherein said given parameters of background acoustic noise include relative noise energy and noise spectrum tilt.

14. (previously presented) The method as defined in claim 10, wherein background acoustic noise in said speech signal is estimated as relative noise energy and noise spectrum tilt, the method further comprising, after said estimating background acoustic noise: deciding whether the estimated relative noise energy for a current speech frame is below a predetermined threshold; and if so, not performing said adapting filter coefficients and applying said filter, and instead per-forming energy attenuation on the current speech frame so as to suppress background acoustic noise in a speech pause.

15. (currently amended) An apparatus comprising:

a filter configured for reduction of distortion, including quantization noise, caused by speech coding of a speech signal;

a noise estimator configured for estimating background acoustic noise in said speech signal; and

a postfilter controller configured for adapting said filter in response to the estimated background acoustic noise, wherein said filter, when applied to said speech signal, reduces background acoustic noise and reduces distortion, including quantization noise, caused by speech coding in said speech signal.

16. (currently amended) The apparatus as in claim 15, wherein said postfilter controller configured for adapting said filter is arranged to adjust filter coefficients of said filter in response to the estimated background acoustic noise.

17. (currently amended) The apparatus as in claim 16, wherein said noise estimator configured for estimating, said postfilter controller configured for adapting and said filter are arranged to operate on portions of said speech signal which contain speech as well as on portions which do not contain speech.

18. (previously presented) The apparatus as in claim 16, wherein said filter includes a short-term filter function designed for attenuation between spectrum formant peaks of said speech signal and wherein said filter coefficients include at least one coefficient that controls the frequency response of said short-term filter function.

19. (currently amended) The apparatus as in claim 15, wherein said noise estimator configured for estimating background acoustic noise is arranged to estimate background acoustic noise as relative noise energy and noise spectrum tilt.

20. (currently amended) The apparatus as in claim 16, wherein said postfilter controller configured for adapting said filter comprises a lookup table, which maps a plurality of values of estimated background acoustic noise to a plurality of filter coefficient values.

21. (currently amended) The apparatus as in claim 15, wherein said speech signal comprises speech frames and wherein said noise estimator configured for estimating, said postfilter controller configured for adapting and said filter are ~~arranged~~ configured to operate on said speech signal on a frame-by-frame basis.

22. (cancelled)

23. (cancelled)

24. (cancelled)

25. (cancelled)

26. (currently amended) A ~~computer readable medium~~memory storing program code, ~~for~~ which when executed ~~execution~~ by a processor ~~for~~ performs: providing a filter suited for reduction of distortion, including quantization noise, caused by speech coding; estimating

background acoustic noise in said speech signal; adapting said filter in response to the estimated background acoustic noise to obtain an adapted filter; and applying said adapted filter to said speech signal so as to reduce background acoustic noise and to reduce distortion, including quantization noise, caused by speech coding in said speech signal.

27. (cancelled)

28. (cancelled)

29. (cancelled)

30. (cancelled)

31. (cancelled)

32. (previously presented) An apparatus comprising:

a filter configured for reduction of distortion, including quantization noise, caused by speech coding of a speech signal;

means for estimating background acoustic noise in said speech signal; and

means for adapting said filter in response to the estimated background acoustic noise, wherein said filter, when applied to said speech signal, reduces background acoustic noise and reduces distortion, including quantization noise, caused by speech coding in said speech signal.